## AMENDMENTS TO THE CLAIMS

A detailed listing of all claims that are, or were, in the present application, irrespective of whether the claim(s) remain(s) under examination in the application is presented below. The claims are presented in ascending order and each includes one status identifier. Those claims not cancelled or withdrawn but amended by the current amendment utilize the following notations for amendment: 1. deleted matter is shown by strikethrough for six or more characters and double brackets for five or less characters; and 2. added matter is shown by underlining.

1. (Cancelled)

2. (Previously Presented) The method according to claim 30, wherein the computing includes:

producing a white acoustic sound signal on the right with an acoustic diffusion system, from a white noise electric signal;

detecting with the acoustic detector a corresponding acoustic signal received in the form of a modified white received electric sound signal on the right and a modified white electric sound signal on the left corresponding to a reception of the white acoustic sound signal on the right;

producing a frequency spectrum on the right corresponding to a white noise electric signal on the right, and two received frequency spectrums, respectively corresponding to the modified white received electric sound signal on the right and to the modified white received electric sound signal on the left;

producing a first set of coefficients from frequency filters from the frequency spectrum on the right and from the frequency spectrum of the modified white received electric sound signal on the right;

producing a second set of coefficients from frequency filters from the frequency spectrum on the right and from the frequency spectrum of the modified white received electric sound signal on the left;

producing a white acoustic sound signal on the left with an acoustic diffusion system, from a white noise electric signal;

detecting a corresponding acoustic signal received in the form of a modified white received electric sound signal on the left and a modified white electric sound signal on the right corresponding to a reception of the white acoustic sound signal on the left with the acoustic detector;

producing a frequency spectrum on the left corresponding to a white noise electric signal on the left, and two received frequency spectrums, respectively corresponding to the modified white received electric sound signal on the left and to the modified white received electric sound signal on the right;

producing a third set of coefficients from frequency filters from the frequency spectrum on the left and from the frequency spectrum of the modified white received electric sound signal on the left;

producing a fourth set of coefficients from frequency filters from the frequency spectrum on the left and from the frequency spectrum of the modified white received electric sound signal on the right, said four sets of coefficients forming a quadrille of coefficient sets; and

filtering the electric sound signals on the right and left with frequency filters whose parameters are given by said quadrille.

3. (Previously Presented) The method according to claim 2, wherein:

the sets of coefficients are produced from the two spectrums by a component to component complex division of complex points from these components in each of these spectrums.

4. (Previously Presented) The method according to claim 2 wherein said computing includes the steps of

producing coefficients of the four temporal filters from coefficients of the first, second, third and fourth frequency filters respectively.

5. (Cancelled)

6. (Previously Presented) The method according to claim 4 wherein the coefficients from a temporal filter whose rank is greater than a given rank are eliminated and wherein the coefficients from a temporal filter whose value is lower than a threshold are eliminated.

7. (Previously Presented) The method according to claim 2 wherein quadrilles of sets of coefficients are produced for different configurations of the acoustic diffusion system and or for different rooms in which the acoustic diffusion system is placed for the production of coefficients.

8. (Previously Presented) The method according to claim 7, wherein one of the configurations is a configuration in a cone of confusion.

9-10. (Cancelled)

11. (Previously Presented) The method according to claim 30 wherein combined electric sound signals on the right and left are filtered on given frequency bands and, a delay is introduced in each of these frequency bands.

12. (Previously Presented) The method according to claim 11, wherein combined electric sound signals on the right and left are filtered by using a high-pass filter, and-high-frequency electric sound signals are obtained, combined electric sound signals on the right and left are filtered by using a low-pass filter, and low-frequency electric sound signals are obtained.

13. (Previously Presented) The method according to claim 12, wherein a first delay is introduced in the low-frequency electric sound signals and a second delay is introduced in the high-frequency electric sound signals.

14. (Previously Presented) The method according to claim 13, wherein the first delay introduced in the low-frequency electric sound signal obtained from the combined electric sound signal on the right is different from the first delay introduced in the low-frequency electric sound signal obtained from the combined electric sound signal on the left, and the second delay introduced in the high-frequency electric sound signal obtained from the combined electric sound signal on the right is different from the second delay introduced in the high-frequency electric sound signal obtained from the combined electric sound signal on the left.

15. (Previously Presented) The method according to claim 30 wherein, to filter,

a signal transform of an electric sound signal is performed and a transformed signal is obtained,

the transformed signal is multiplied by filtering coefficients and a multiplied signal is obtained,

the multiplied signal is transformed by an inverse transform, and

the filtering coefficients are coefficients of finite impulse response filters.

16. (Previously Presented) The method according to claim 15, wherein, to perform the transform

a frame of the electric sound symbol is divided into N blocks,

the transform of each of the blocks is performed,

the filtering coefficients are divided into N packets of coefficients,

the N blocks of input data are multiplied two by two by the N packets of filter coefficients, and

the multiplied blocks are added to obtain the multiplied signal.

17. (Previously Presented) The method according to claim 16, wherein to divide the frame and to calculate the transform,

the transform of each of the N blocks is calculated successively, and

the transformed blocks are transmitted to a delay line at N outputs.

18. (Previously Presented) The method according to claim 16 wherein, to divide the frame into N blocks,

an electric sound signal is stored in a circular buffer memory with capacity proportional to the nth of the frame of the electric sound signal.

19. (Previously Presented) The method according to claim 16 wherein,

to divide a frame of the signal into N blocks, double blocks are formed that are overlayed on each other by half,

the transform of each of the double blocks is performed,

the N packets of coefficients are completed by the constant samples to obtain double packets,

each of the N double blocks are multiplied by one of the N double packets and multiplied double blocks are obtained, and

the multiplied blocks are extracted from the multiplied double blocks.

20. (Previously Presented) The method according to claim 30 wherein, to compute,

an artificial head that comprises two acoustic detectors is placed in a median axis of two acoustic diffusion systems,

an electric signal in the form of a Dirac comb is applied simultaneously as input to the two acoustic diffusion systems, and

these direct fields and these crossed fields received by the acoustic detectors are aligned two by two by varying the position of the artificial head.

21. (Previously Presented) The method according to claim 30 wherein, to diffuse,

equalization functions are incorporated in the cells situated upstream from the Fourier transform cells.

22. (Previously Presented) The method according to claim 21, wherein

the frequency components of four frequency filters obtained from the four modified temporal filters are adjusted independently.

23. (Previously Presented) The method according to claim 30 wherein, to diffuse,

at least one of a phase and an amplitude of temporal filter coefficients are modified along all or part of an impulse response.

24. (Previously Presented) The method according to claim 15, wherein, to perform the transform,

the filtering temporal coefficients are divided into Q slots (HDD1-HDD4) of coefficients with progressive length M, 2M, 4M,...(2^(Q-1))M points,

the transform of each of these slots is performed and transformed slots are obtained,

a frame of the electric sound signal is divided into blocks (x1-x8) with a length of M points,

the transform of each of these blocks is performed and transformed blocks are obtained, and

the transformed blocks are multiplied by the transformed slots and corresponding multiplied blocks are obtained by inverse transformation to the blocks of signals that half-overlap each other two by two in time.

25. (Previously Presented) The method according to claim 24 wherein, to perform the inverse transformations of multiplied blocks,

a first multiplied block with a length of 2P x M points, a temporal block corresponding in time to this first multiplied block, a second multiplied block corresponding in time to a second temporal block are modulated , this first and second temporal block are overlayed by half in time, and

a modulated block with a length of 2P x M points is obtained, then

this modulated block with a length of 2P x M points is added to the second block, and

a combined block with a length of 2P x M points is obtained.

26. (Previously Presented) The method according to claim 25, wherein, to modulate,

the odd components of a multiplied block with a length of 2M points wherein the block corresponding to it in time is overlayed with another is multiplied by –1, and the even components are multiplied by +1.

27. (Previously Presented) The method according to claim 25 wherein, to perform the inverse transformations of multiplied blocks with a length of 2M points,

the even components of the combined block with a length of 2P x M points are selected, and

an even block with a length of 2(P-1) x M points is obtained

this even block is multiplied by 1/2 and the result of this multiplication is added to an auxiliary multiplied block with a length of 2(P-1) x M points, and

a compensation block is obtained.

28. (Previously Presented) The method according to claim 25 wherein to perform the inverse transformations of multiplied blocks with a size of (2P)M,

the odd components of the combined block with a size of 2P x M points are selected, and

an odd block with a length of 2(P-1) x M points is obtained,

an inverse transform of this odd block with a length of (2(P-1))M points is performed, and

an odd inversed block is obtained that is situated in a temporal domain, then

the odd inversed block is multiplied by a complex coefficient conjugated from a complex coefficient W(n), and

an odd normalized inversed block with a length of 2(P-1) x M points is obtained.

29. (Previously Presented) The method according to claim 30, wherein a time lag is introduced between the original electric sound signals and the processed electric sound signals.

30. (Currently Amended) A method for processing an electric sound signal wherein a right sound signal and a left sound signal are diffused in a reflective environment by two speakers and are detected by an acoustic detector comprising a right microphone and a left microphone, the method comprising:

computing a first temporal filter ~~corresponding to a detection by~~ representing a first acoustic transformation applied to the right sound signal by the reflective environment between the right speaker and the right microphone ~~of the right sound signal~~;

computing a second temporal filter ~~corresponding to a detection by~~ representing a second acoustic transformation applied to the right sound signal by the reflective environment between the right speaker and the left microphone ~~of the right sound signal~~;

computing a third temporal filter ~~corresponding to a detection by~~ representing a third acoustic transformation applied to the left sound signal by the reflective environment between the left speaker and the left microphone ~~of the left sound signal~~;

computing a fourth temporal filter ~~corresponding to a detection by~~ representing a fourth acoustic transformation applied to the left sound signal by

the reflective environment between the left speaker and the right microphone ~~of the left sound signal~~;

modifying each of the temporal filters by an operation including at least one of:

normalizing the temporal filters on a maximum of a direct field or on a quadratic average,

temporal resetting of the temporal filters in relation to each other,

providing a time lag of samples from a temporal filter,

masking of at least some of the samples from the temporal filter, and

altering an amplitude of at least some of the samples from a temporal filter;

applying the modified temporal filters to a right original sound signal and a left original sound signal to obtain processed electric sound signals by:

applying a first modified temporal filter to the right original electric sound signal to obtain a first processed electric sound signal,

applying a second modified temporal filter to the right original electric sound signal to obtain a second processed electric sound signal,

applying a third modified temporal filter to the left original sound signal to obtain a third processed electric sound signal, and

applying a fourth modified temporal filter to the left original sound signal to obtain a fourth processed electric sound signal,

adding the first and fourth processed electric sound signals and the right original sound signal to obtain a right processed electric sound signal;

adding the second and third processed electric sound signals and the left original sound signal to obtain a left processed electric sound signal; and

diffusing the right processed electric sound signal and the left processed sound signal.